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PROVISIONAL SPECIFICATION

Invention title: **Adaptive Directional Systems**

The invention is described in the following statement:

chbm M0111431318v1 305265977

Adaptive Directional Systems

Field of the invention

The invention relates to adaptive directional systems, and more particularly to a method and apparatus for producing adaptive directional signals. The invention may be applied to the provision of audio frequency adaptive directional microphone systems for devices such as hearing aids and mobile telephones.

Background of the invention

In this specification, where a document, act or item of knowledge is referred to or discussed, this reference or discussion is not an admission that the document, act or item of knowledge or any combination thereof was at the priority date:

(i) part of common general knowledge; or

(ii) known to be relevant to an attempt to solve any problem with which this specification is concerned.

An omni-directional microphone converts sound waves emanating from all directions into electrical signals to be passed to an output. A directional microphone system is typically constructed from two or more omni-directional microphones, in a configuration that attenuates sounds emanating from certain directions and enhances sounds emanating from other directions.

The directionality of a particular directional microphone system is represented graphically by a polar pattern, where the direction directly in front of the microphone is shown at 0°, and the direction directly behind the microphone is shown at 180°. The plot of a polar pattern represents gain as a function of the direction of sound arrival, the gain for any given direction represented by the distance from the centre of the polar coordinates.

Some of the more common polar patterns are illustrated in Figure 1, which shows an omni-directional polar pattern 10 (with no nulls), a bi-directional polar pattern 12 (with nulls at 90° and 270°), a cardioid polar pattern 14 (with a null at 180°) and a super-cardioid polar pattern 16 (with nulls at approximately 135° and 225°)

Directional microphone systems have been employed in the past in hearing aids to improve the signal-to-noise ratio. It is assumed the sound that the listener wishes to hear emanates from a forward direction, ie the direction in front of the listener, and so the directional microphone system is designed to provide a maximum gain for sounds

emanating from this direction whilst attempting to reduce the sounds emanating from other directions.

Conventionally, directional microphone systems are fixed, meaning that the output signal has a fixed polar pattern. Fixed directional microphones traditionally comprise two spaced omni-directional microphones, a delay element and a difference element, and are configured to provide a fixed directional signal by subtracting the delayed signal from the original signal.

Examples of fixed directional microphone systems that do not utilise a delay element are disclosed in US-5,463,694 and US-4,712,244. These directional system instead use a particular combination of averaging, amplifying, summing, subtracting and integrating elements that operate on the signals from the microphones to construct the fixed directional signal pattern.

As the output from a fixed directional microphone system is a polar pattern with a stationary null, it can only maximally attenuate sounds emanating from a particular direction (although sounds from directions close to the null will receive some attenuation). In many practical situations this can represent a significant compromise on the performance of the system. If noise emanates from a direction different to that of the null, or from multiple directions (which would require a compromise null position), or if there is a moving noise source, a reduced signal-to-noise ratio will result.

More complex 'adaptive' directional microphone systems have been developed to overcome shortcomings in directional microphone systems. Such systems have the ability to construct varying polar patterns which are able to dynamically 'steer' a null to attenuate signals representing sounds emanating from different directions, or from moving sources.

Known adaptive directional microphone systems are in fact extensions of conventional fixed systems, and utilise a variable delay element to vary the polar patterns, and thus provide adaptive directional signals. The architecture of such an adaptive directional microphone system is illustrated in Figure 2. Front 20 and rear 22 omni-directional microphones transduce sound waves (not illustrated) into front 21 and rear 23 electrical signals.

When a sound wave arrives from the forward direction, it reaches the front microphone first, and hence the rear signal 23 is a delayed version of the front signal 21. Likewise, if the sound arrives from behind, the front signal 21 is a delayed version of the rear signal 23. If the sound arrives from the side, there is no delay between the two signals 21 and 23. In short, the delay between the two signals is dependent on the angle of arrival of the

sound wave. A variable delay element 24, coupled to the rear microphone 22, is used to match the delay corresponding to the desired cancellation direction. This produces a delayed rear signal 25. This signal 25 is received by a difference element 26 also coupled to the front microphone 20, configured as shown to output the difference between signals 21 and 25 to produce the directional output signal 30. As will be understood by those skilled in the art, the adaptive nature of this system is provided by a feedback loop, the adaptive directional signal 30 feeding back to an optimising algorithm element 28, which in turn provides an optimised delay value 29 to the variable delay element 24 used in producing delayed rear signal 25. The system is therefore designed to iteratively converge to a desired solution, in accordance with the algorithm implemented by element 28.

Examples of known adaptive directional microphone systems that use variable delay elements are described in US-5,757,933, US-2001/0028720, US-2001/0028718, US-6,539,096 and US-6,339,647. The main disadvantages of these systems are the complexity involved in implementing the variable delay element, along with the possible instability introduced through the use of a feedback structure.

Adaptive directional microphone systems that do not employ variable delay elements are also known, and examples of such systems are described in WO-01/97558 and US-2003/0031328. Both systems utilise two *fixed* delay elements to generate a forward-facing and a backward-facing cardioid polar pattern, which respectively represent an 'enhanced signal' and an 'enhanced noise'. The enhanced noise and enhanced signal are then combined to produce an adaptive directional signal. An optimisation algorithm is used to find the ideal combination of the two signals to give maximum noise rejection. A major disadvantage of these adaptive directional systems is again their reliance on delay elements, in this case multiple fixed delay elements. As discussed above, these elements can be very difficult to implement in hardware, or require a specially designed allpass filter, which significantly increases the processing requirements of the system, particularly when implemented using a digital signal processor.

Adaptive directional microphone systems have also been developed that, instead of being continuously variable, simply select an output from a range of signals that have been implemented. One of the simplest approaches is described in US-6,327,370, and involves using a fixed directional signal and an omni-directional signal, with a selection between the signals based on prescribed criteria such as ambient noise level. The idea has been extended in the teaching of US-6,522,756, which includes a greater number of directional signals for selection. Such 'signal selection' systems are quite simple and can perform

well, however for adequate performance they require many signals to be generated simultaneously, greatly increasing the demands on hardware and processing power. In addition, the limited choice of beam types signifies a discontinuous response, such that a signal with an optimum polar pattern cannot always be found.

There remains a need to provide an improved, or at least an alternative, method and apparatus for producing adaptive directional signals.

Summary of the invention

According to one form of the invention a method for producing an adaptive directional signal is provided, the method including the step of constructing the adaptive directional signal from a weighted sum of a first signal having an omni-directional polar pattern and a second signal having a bi-directional polar pattern, wherein the weights are calculated to give the combined signal a constant gain in a predetermined direction and to minimise the power of the constructed adaptive directional signal.

By minimising the power of the constructed adaptive directional signal, the amplitude of signals not received from the predetermined direction is minimised.

The directional signal is produced by the optimised weights that in effect, adaptively vary the relative contributions of the first and second signals, to thereby minimise or eliminate the contribution of signals emanating from directions other than the predetermined direction. Thus it will be realised that the polar pattern of the combined signal will vary in response to changes in the first and second signals, whilst providing a constant gain for signals that emanate from the predetermined direction. For example, the adaptive directional signal may have a cardioid, super-cardioid, or even an omni-directional polar pattern.

In a preferred embodiment, the first and second signals are derived from signals produced by two spaced omni-directional microphones, a front and a rear microphone, and said predetermined direction is the forward direction along the microphone axis. The method of the present invention is also applicable to signals produced from an array of more than two microphones.

Preferably, the second signal is provided by the difference between signals produced by two spaced omni-directional microphones, without the use of a delay element.

In accordance with this embodiment, a further step may be included of processing the second signal by means of an integrator element or an integrator-like filter before constructing the combined signal, thereby compensating for the attenuation of low

frequencies and phase shifts introduced in the subtraction of the two omni-directional signals.

The method may also include the step of amplifying the second signal after the step of integrating the second signal and before the step of constructing the adaptive directional signal, to ensure an equivalent forward gain between the second signal and the first signal.

Preferably, the microphones are matched, which can be accomplished by using physically matched microphones or by employing a gain element to match the microphone outputs.

A weight may be calculated in any convenient manner that provides for the constant gain of the combined polar pattern in the forward direction and minimises the power of the combined signal. Typically the constant gain is provided by imposing a constraint that the first signal weight and the second signal weight add to one.

In preferred embodiments the weights are calculated in a non-iterative manner, such as by solving the following equation:

$$a = \frac{\sum y^2 - \sum xy}{\sum x^2 - 2\sum xy + \sum y^2}$$

Where:

a = weight for the first signal

(1-a) = weight for the second signal

x = first signal sample

y = second signal sample.

A weight may be calculated for a frame of predetermined length consisting of N first signal samples and N second signal samples. The length of the frame (N) generally depends upon the environment of application of the method, however a suitable frame length for audio frequency signals is 64 samples long. The weighting factor may change significantly from frame to frame, so the series of weight values may also be filtered or smoothed to minimise frame to frame variation in the weight (which may otherwise be heard as audible artifacts).

In another embodiment weights are calculated continuously for each first signal sample and second signal sample. This is achieved by calculating x^2, y^2 and xy for each sample and adding them to the appropriate running sum. A leaky integrator (an integrator having a feedback coefficient slightly less than one) can be used to perform the running sum in

order to prevent overflows and to ensure that the systems "memory" is not too long. This embodiment allows a new weighting factor to be calculated every time that a new sample is available, rather than having to wait for a whole frame of samples.

In another embodiment the variables x and y can be frequency domain samples rather than time domain samples. In this case the optimization of the weighting factor 'a' can be calculated as above, but with the added advantage that the weighting factor can be calculated and applied to several independent subsets of frequency domain samples (giving different directional responses at different frequencies). Also, if some frequencies are deemed to be more important to suppress than others, they can be given a higher weighting before calculating the weighting factor 'a'.

The sums used for calculating the weighting factor 'a' can also be used to detect particular conditions that require a different strategy. For example, if Σx^2 is particularly small, then the environment is quiet, which suggests that an omni-directional response is more suitable than a directional response. In this case a simple threshold test could be performed to decide on the appropriate strategy.

The invention is based on the realisation that an adaptive directional signal of varying polar pattern can be constructed from a weighted sum of an omni-directional and a bi-directional polar pattern which can be easily generated without the use of delay elements. Surprisingly, despite the simplicity of the system of the invention, theoretical analysis and test results have demonstrated extremely good performance in terms of noise reduction and signal enhancement.

According to a further form of the invention an apparatus for producing an adaptive directional signal is provided, the apparatus including:

means for producing a first signal having an omni-directional polar pattern and a second signal having a bi-directional polar pattern; and

means for constructing the adaptive directional signal from a weighted sum of the first and second signals, wherein the weights are calculated to give the combined signal a constant gain in a predetermined direction and to minimise the power of the constructed adaptive directional signal.

Preferably the apparatus further includes means for integrating (or filtering with an integrator-like filter) the second signal before constructing the adaptive directional signal, thereby compensating for attenuation of low frequencies and phase shifts introduced in the production of the second signal.

The invention thus serves to provide a directional response that adaptively provides the desired performance, by fixing the gain in the forward direction, while minimising the power received.

Importantly, and in contrast with the prior art, the invention avoids the need to use delay elements in providing the adaptive directional response. Instead of an iterative approach converging on a desired solution, the method of the present invention mathematically calculates the required weights to apply to combining the signal patterns in accordance with the preset constraints on a frame-by-frame or sample-by-sample basis.

The invention can also be applied to sub-band processing, providing a different adaptive response in different frequency bands.

Description of the drawings

The invention will now be further explained and illustrated by way of a non-limiting example and with reference to the accompanying drawings, in which:

Figure 1 is an illustration of the polar patterns of various directional signals;

Figure 2 is a schematic drawing of an adaptive directional microphone system of the prior art;

Figure 3 is a schematic drawing of an apparatus for producing an adaptive directional signal in accordance with an embodiment of the present invention;

Figure 4 is a flow chart representing of a method for producing an adaptive directional signal in accordance with an embodiment of the present invention; and

Figure 5 illustrates two example adaptive directional signals produced by implementing the method of the present invention.

Turning to Figure 3, the architecture of an apparatus for producing an adaptive directional signal is illustrated. The same reference numerals as those used in Figure 2 are employed to reference similar components. The apparatus is configured as explained below to combine the output of multiple microphones to produce an adaptively directional output. Front 20 and rear 22 omni-directional microphones respectively transduce sound waves into front 21 and rear 23 signals. Microphones 20 and 22 should be matched, and this can be accomplished either by using physically matched microphones or by employing a gain element (shown at 35 in Figure 3) to match the microphone outputs. The front 20 and rear 22 microphones also include suitable analogue-to-digital converters (not shown) for providing the front 21 and rear signals 22 in a digital form.

As noted above, the delay between the front signal and the rear signal will depend on the angle that the sound arrives from. Front signal 21 and rear signal 23 are passed to a differencing element 26 for subtraction of rear signal 23 from front signal 21 to produce a signal 34 with a bi-directional polar pattern. This bipolar signal 34 attenuates sounds emanating from directions perpendicular to the axis of the front 20 and rear 22 microphones, whilst front signal 21 retains an omni-directional polar pattern.

Because the bi-directional signal 34 is generated by the difference between two delayed samples it inherently introduces a differentiated (high pass) frequency response that tends to produce undesirable attenuation of lower frequencies and a phase shift at all frequencies. To counter this effect, the bi-directional signal 34 is passed to an integrator in order give the signal 34 a flat frequency response and at the same time to automatically correct for the phase shift that is introduced during construction of the bi-directional signal. This integrator can also be replaced by a filter with a similar (but not identical) response to the integrator. This allows other undesirable artifacts (such as a dc offset) to be removed from the bi-directional signal.

The integrated signal 36 and the front microphone (omni-directional) signal 21 are directed to an optimiser 38 that calculates respective front signal weights 39A and rear signal weight 39B by means of an optimising algorithm described in further detail below.

The optimiser 38 calculates weights 39A and 39B subject to the constraint that the directional response of the system has a constant gain in the forward direction. Where the signals are of audio frequency and the system is employed in a hearing aid, this direction will generally be selected as the forward direction, ie, along the axis of the front 20 and rear 22 microphones. This is in accordance with the assumption noted above that the listener wishes to hear sounds emanating from the forward direction.

The constant gain in the forward directional is achieved by constraining the weights 39A and 39B to add to 1.0. This prevents sound emanating from the forward direction being attenuated in the adaptive directional signal produced by the apparatus.

It should be realised however, that the weights can be calculated to give a constant gain to signals emanating from a selected other direction, which may be useful in other applications or in accordance with other microphone configurations.

The optimisation algorithm is configured to calculate weights 39A and 39B to minimise the signal power produced. By minimising the power of the signal, the noise component (defined as signals from any direction other than the front) is minimised, thereby providing an improved signal-to-noise ratio.

The weights 39A and 39B calculated by the optimiser 38 in accordance with the optimisation algorithm are applied to respective variable gain elements 40A and 40B to which front signal 21 and bi-directional signal 36 are passed. The variable gain elements thus apply weighted gains to the samples that comprise signals 21 and 36, to produce respective weighted signals 42A and 42B.

The weighted signals 42A and 42B are then passed to a summing element 44 that outputs an adaptive directional signal 46 by summing the weighted signals 42A and 42B. The adaptive directional signal 46 is then processed further (if required) and then output to suitable output means, such as an earphone speaker (not shown).

Turning to Figure 4, the steps carried out by the optimiser in calculating the weights are illustrated with reference to a flow chart. In use, the optimiser is a suitable digital signal processing apparatus, as would be understood by those skilled in the art. At steps 50 and 52 the optimiser receives a sampled value of the omni-directional signal and the bi-directional signal. In this embodiment, the weights are calculated on a frame by frame basis, with each frame being 64 samples long. Therefore, at step 56 a test is performed of whether the end of the frame has been reached. If the test is negative, step 54 is carried out and the value of the omni-directional sample and bi-directional sample are accumulated in the following summations:

$\sum x^2$

$\sum y^2$ and

$\sum xy$

where x= the omni-directional sample series; and

y= the bi-directional sample series.

If the test is positive, the weight for the omni-directional signal 'a' is calculated using the accumulated sums in the following formula:

$$a = \frac{\sum y^2 - \sum xy}{\sum x^2 - 2\sum xy + \sum y^2}$$

As noted above, the weight is optimised subject to the constraint that there is to be a constant gain in the forward direction, which is imposed by setting the omni-directional and bi-directional weights equal to one. From this, the bi-directional weight is simply calculated as (1-a). Also, as noted previously, other criteria can be applied in calculating

'a', such as forcing it to 1 (i.e. an omni-directional response) when in a quiet environment (if $\sum x^2$ is small).

The derivation of the above formula is found by using the constraint that the total power of the output adaptive directional signal is to be minimised. Therefore:

$$Energy = \sum (ax(t) + (1-a)y(t))^2$$

Differentiating with respect to 'a' to find the point of minimum energy gives:

$$\frac{dEnergy}{da} = 0$$
$$= 2a\left(\sum x^2 - 2\sum xy + \sum y^2\right) + 2\left(\sum xy - \sum y^2\right)$$

Solving for 'a' gives:

$$a = \frac{\sum y^2 - \sum xy}{\sum x^2 - 2\sum xy + \sum y^2}$$

Returning to the flow chart at step 60, the calculated weights are filtered to guard against excessive frame to frame variation in the weights.

In an alternative embodiment, the values Σx^2, Σy^2 and Σxy are filtered prior to their use in calculating the weights. This can be particularly useful when processing samples continuously and can be implemented efficiently if the summing operations used in the calculations of the weights are implemented as 'leaky integrators' (ie an integrator with a feedback coefficient slightly less than one). This allows a new weighting factor to be calculated every time a new sample is available, rather than having to wait for a whole frame of samples. The final step 62 in the process illustrated is the outputting of the weights 42A and 42B.

In a further alternative embodiment the weights may be calculated over multiple frames, or continuously.

Turning to Figure 5, the effect of different omni-directional and bi-directional weights on the polar pattern of the output adaptive directional signal produced (under the constraints defined above) is illustrated. The directional signal (46 and 46' in Figure 5) is constructed from the weighted contributions of the omni-directional 42A/42A' and bi-directional signals 42B/42B'.

For example, an omni-directional weight of 0.5 and a bi-directional weight of 0.5 produce a directional signal 46 having a cardioid polar pattern as shown. The equal weighting

used means that the rear lobe of the bi-directional signal exactly cancels with the omni-directional signal in that direction.

In the second example in Figure 5, the omni-directional signal 42A' and bi-directional signal 42B' are given weights of 0.375 and 0.625 respectively, providing a directional signal having a super-cardioid polar pattern as illustrated.

It should also be noted that in certain situations, due to the constraints imposed in accordance with the invention, an adaptive directional signal having an omni-directional polar pattern may be produced, ie when an omni-directional weight of 1 (and thus a bi-directional weight of 0) is applied. This can be the result, for example, in quiet conditions or in conditions with high levels of wind noise. In such situations the omni-directional pattern is desirable, and in contrast with prior art systems (which require to be be configured to switch to an omni-directional pattern under prescribed conditions), the invention allows the system to automatically adopt such a response.

The invention can be implemented in hardware or software, and in the application to a hearing aid is preferably implemented in a DSP chip, with samples from the signals produced by each microphone used to calculate the fixed polar patterns employed as inputs to the adaptive directionality process. Modifications and improvements to the invention will be readily apparent to those skilled in the art. Such modifications and improvements are intended to be within the scope of this invention. For example, whilst the above has been described by reference to the time domain, the teachings of the present invention apply equally in the frequency domain.

Dynamic Hearing Pty Ltd 1 December, 2003

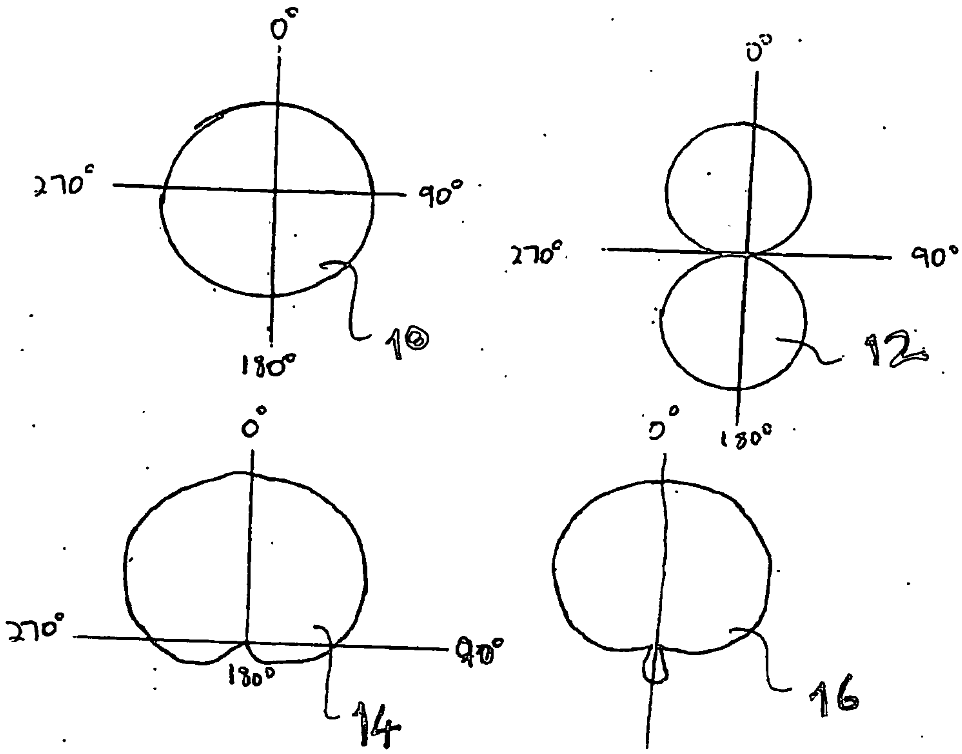

Fig. 1

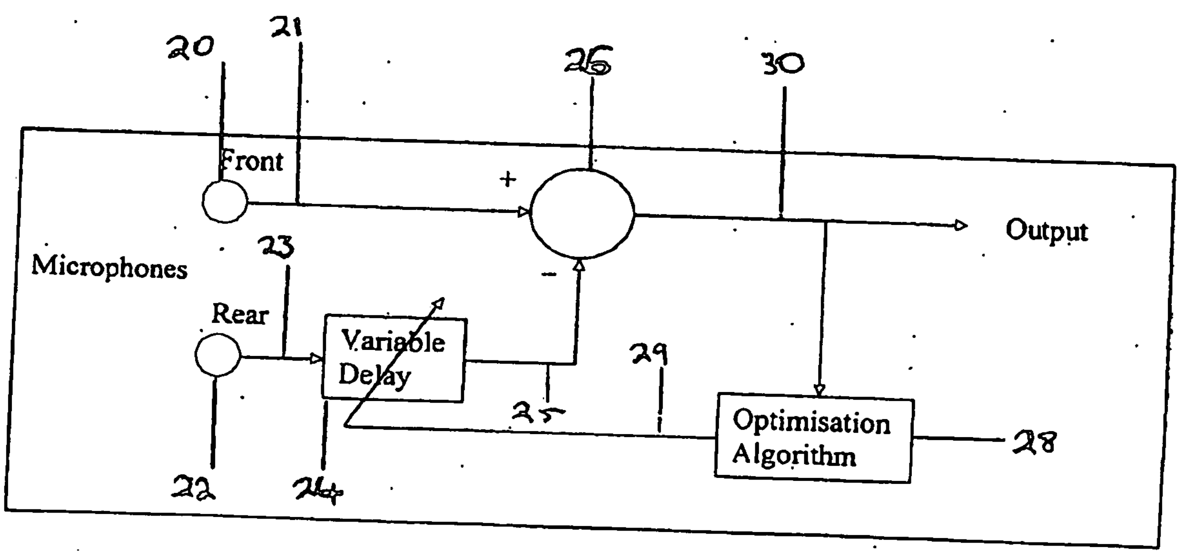

Fig. 2 PRIOR ART

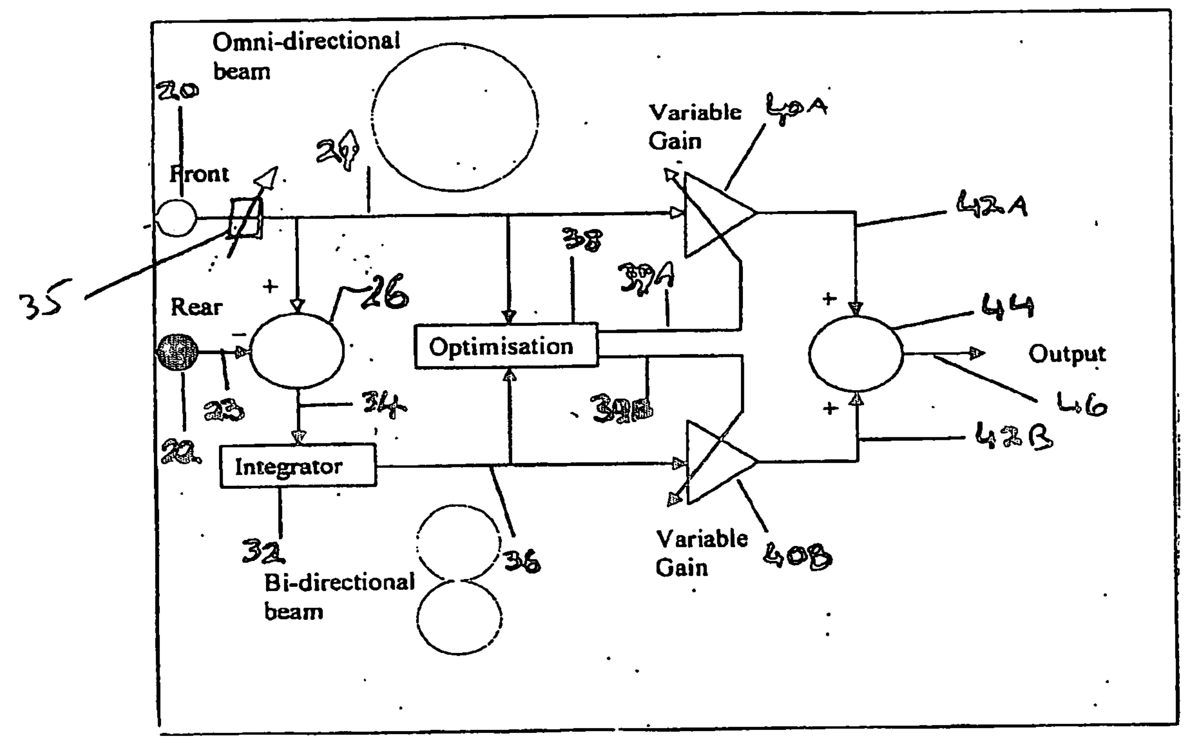
Omni-directional beam
20
Front
24
35
Rear
26
22
23
34
Integrator
32
Bi-directional beam
Optimisation
38
39A
39B
36
Variable Gain
40A
Variable Gain
40B
42A
42B
44
Output
46

Fig. 3

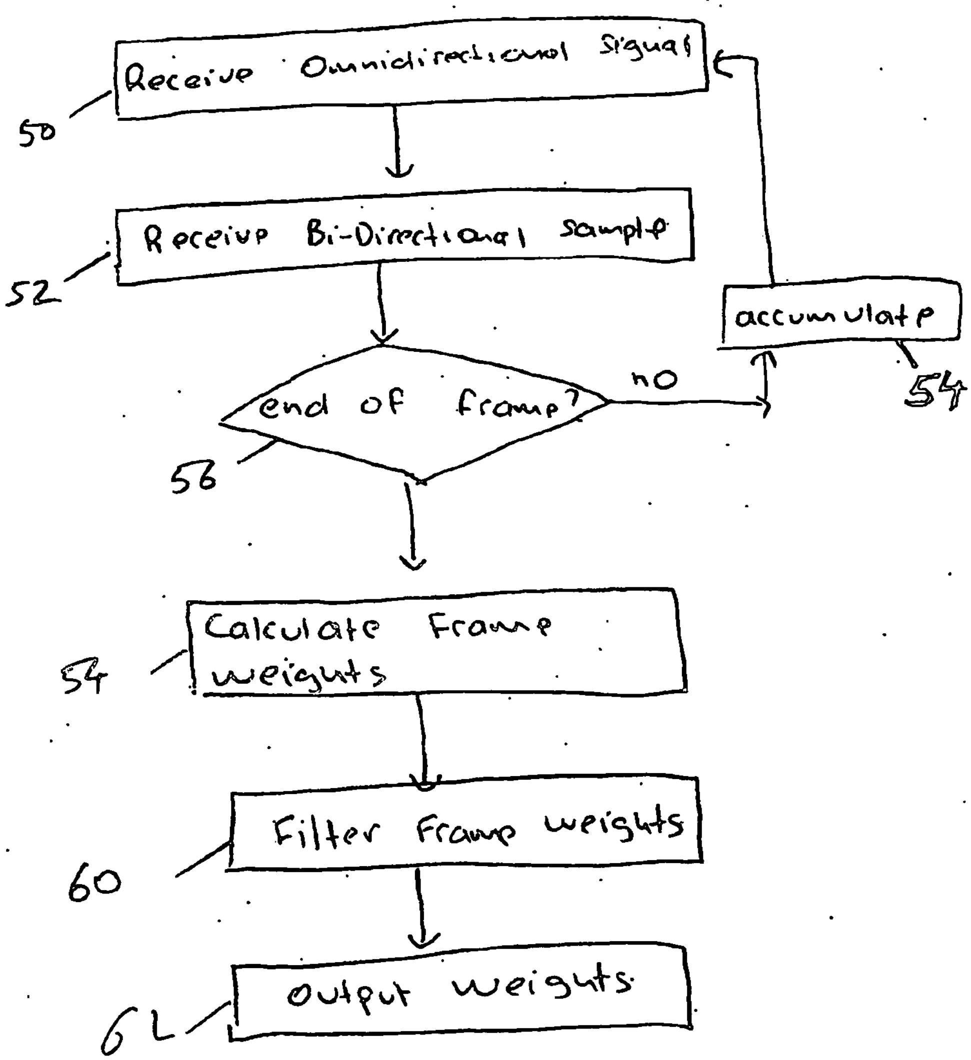
Receive Omnidirectional Signal
50
Receive Bi-Directional sample
52
accumulate
54
end of frame?
no
56
Calculate Frame weights
54
Filter Frame weights
60
output weights
62

Fig.. 4.

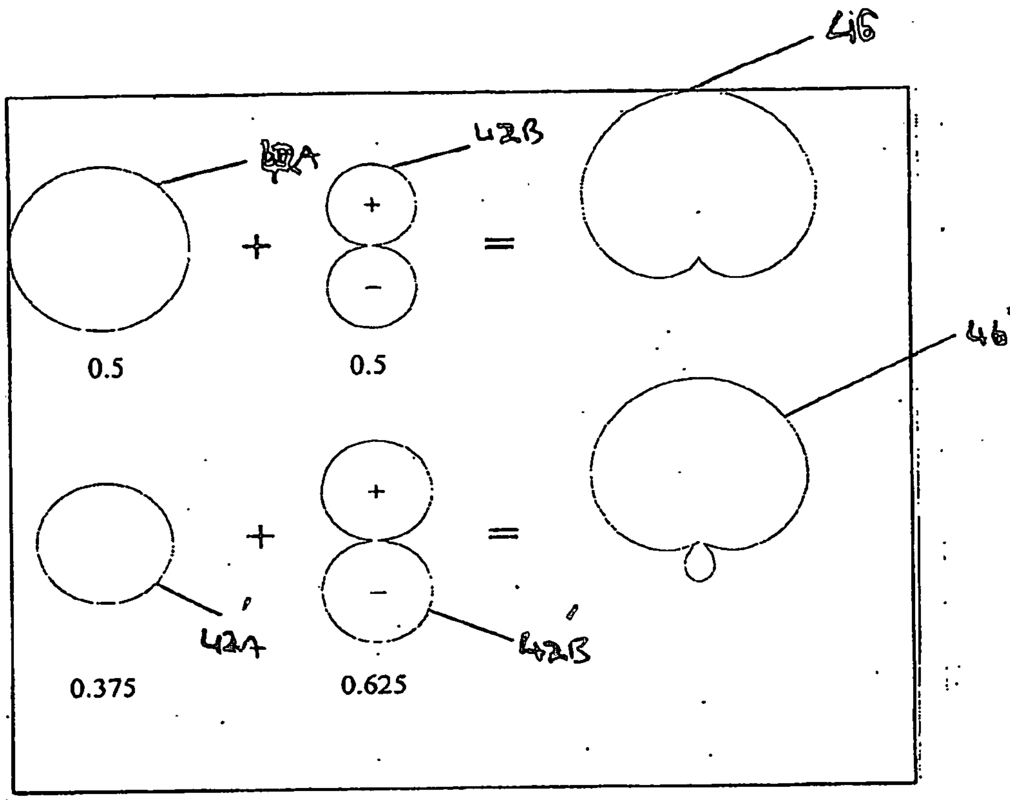

Fig. 5